JC17 Rec'd PCT/PTO 14 MAY 2001

FORM PTO-1390 (Modified) (REV 11-2000) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE

TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371

ATTORNEY'S DOCKET NUMBER
1279-277

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR
09/831843

INTERNATIONAL APPLICATION NO.	INTERNATIONAL FILING DATE	PRIORITY DATE CLAIMED
PCT/US99/28449	**01 December 1999 (01.12.99)**	**01 December 1998 (01.12.98)**

TITLE OF INVENTION
ENHANCED WAVEFORM INTERPOLATIVE CODER

APPLICANT(S) FOR DO/EO/US
Oded GOTTESMAN

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)). The submission must include itens (5), (6), (9) and (24) indicated below.
4. ☐ The US has been elected by the expiration of 19 months from the priority date (Article 31).
5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))
   - a. ☐ is attached hereto (required only if not communicated by the International Bureau).
   - b. ☒ has been communicated by the International Bureau.
   - c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☐ An English language translation of the International Application as filed (35 U.S.C. 371(c)(2)).
   - a. ☐ is attached hereto.
   - b. ☐ has been previously submitted under 35 U.S.C. 154(d)(4).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))
   - a. ☒ are attached hereto (required only if not communicated by the International Bureau).
   - b. ☐ have been communicated by the International Bureau.
   - c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   - d. ☐ have not been made and will not be made.
8. ☐ An English language translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).
10. ☐ An English language translation of the annexes of the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).
11. ☐ A copy of the International Preliminary Examination Report (PCT/IPEA/409).
12. ☒ A copy of the International Search Report (PCT/ISA/210).

Items 13 to 20 below concern document(s) or information included:

13. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
14. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
15. ☐ A **FIRST** preliminary amendment.
16. ☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.
17. ☐ A substitute specification.
18. ☐ A change of power of attorney and/or address letter.
19. ☐ A computer-readable form of the sequence listing in accordance with PCT Rule 13ter.2 and 35 U.S.C. 1.821 - 1.825.
20. ☐ A second copy of the published international application under 35 U.S.C. 154(d)(4).
21. ☐ A second copy of the English language translation of the international application under 35 U.S.C. 154(d)(4).
22. ☒ Certificate of Mailing by Express Mail
23. ☐ Other items or information:

PCTUS1/REV03

JC18 Rec'd PCT/PTO 14 MAY 2001

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/831843	INTERNATIONAL APPLICATION NO. PCT/US99/28449	ATTORNEY'S DOCKET NUMBER 1279-277

24. The following fees are submitted:.

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

CALCULATIONS | PTO USE ONLY

- ☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO and International Search Report not prepared by the EPO or JPO **$1000.00**
- ☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO **$860.00**
- ☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO **$710.00**
- ☒ International preliminary examination fee (37 CFR 1.482) paid to USPTO but all claims did not satisfy provisions of PCT Article 33(1)-(4) **$690.00**
- ☐ International preliminary examination fee (37 CFR 1.482) paid to USPTO and all claims satisfied provisions of PCT Article 33(1)-(4) **$100.00**

				Calculations	PTO use only
ENTER APPROPRIATE BASIC FEE AMOUNT =				**$690.00**	
Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)).				**$0.00**	
CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE		
Total claims	34 - 20 =	14	x $18.00	**$252.00**	
Independent claims	8 - 3 =	5	x $80.00	**$400.00**	
Multiple Dependent Claims **(check if applicable).**			☐	**$0.00**	
TOTAL OF ABOVE CALCULATIONS =				**$1,342.00**	
☒ Applicant claims small entity status. (See 37 CFR 1.27). The fees indicated above are reduced by 1/2.				**$671.00**	
SUBTOTAL =				**$671.00**	
Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)). +				**$0.00**	
TOTAL NATIONAL FEE =				**$671.00**	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable).** ☐				**$0.00**	
TOTAL FEES ENCLOSED =				**$671.00**	
				Amount to be: refunded	$
				charged	$

a. ☐ A check in the amount of ________ to cover the above fees is enclosed.

b. ☒ Please charge my Deposit Account No. **50-0337** in the amount of **$671.00** to cover the above fees. A duplicate copy of this sheet is enclosed.

c. ☒ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. **50-0337** A duplicate copy of this sheet is enclosed.

d. ☐ Fees are to be charged to a credit card. **WARNING:** Information on this form may become public. **Credit card information should not be included on this form.** Provide credit card information and authorization on PTO-2038.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

BERLINER, Robert
FULBRIGHT & JAWORSKI L.L.P.
865 S. Figueroa Street, 29th Floor
Los Angeles, California 90017-2571

SIGNATURE

Robert Berliner
NAME

20,121
REGISTRATION NUMBER

May 14, 2001
DATE

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09/831843

TRANSMITTAL LETTER TO THE UNITED STATES RECEIVING OFFICE

Date	May 14, 2001
International Application No.	PCT/US99/28449
Attorney Docket No	1279-277

I. Certification under 37 CFR 1.10 (if applicable)

Express Mail mailing number	Date of Deposit
EL136019823US	14 May 2001

I hereby certify that the application/correspondence attached hereto is being deposited with the United States Postal Service "Express Mail Post Office to Addressee" service under 37 CFR 1.10 on the date indicated above and is addressed to Assistant Commissioner for Patents, Washington, D.C. 20231.

Signature of person mailing correspondence	Typed or printed name of person mailing correspondence
Linda J. Werk	Linda J. Werk

II. ☒ New International Application

TITLE	ENHANCED WAVEFORM INTERPOLATIVE CODER

Earliest priority date (Day/Month/Year)
01 December 1998

SCREENING DISCLOSURE INFORMATION: In order to assist in screening the accompanying international application for purposes of determining whether a license for foreign transmittal should and could be granted and for other purposes, the following information is supplied. (Note: check as many boxes as apply):

A. ☐ The invention disclosed was **not** made in the United States.

B. ☒ There is no prior U.S. application relating to this invention.

C. ☐ The following prior U.S. application(s) contain subject matter which is related to the invention disclosed in the attached international application. *(NOTE: priority to these applications may or may not be claimed on form PCT/RO/101 (Request) and this listing does not constitute a claim for priority).*

application no.		filed on	
application no.		filed on	

D. ☐ The present international application ☒ is identical ☐ contains less subject matter than that found in the prior U.S. application(s) identified in paragraph C.

E. ☐ The present international application ☐ contains additional subject matter not found in the prior U.S. application(s) identified in paragraph C. above. The additional subject matter is found on pages ________ and ☐ **DOES NOT ALTER** ☐ **MIGHT BE CONSIDERED TO ALTER** the general nature of the invention in a manner which would require the U.S. application to have been made available for inspection by the appropriate defense agencies under 35 U.S.C. 181 and 37 CFR 5.1. See 37 CFR 5.15

III. ☐ A Response to an Invitation from the RO/US. The following document(s) is (are) enclosed:

A. ☐ A Request for An Extension of Time to File a Response

B. ☐ A Power of Attorney (General or Regular)

C. ☐ Replacement pages:

pages		of the request (PCT/RO/101)	pages		of the figures
pages		of the description	pages		of the abstract
pages		of the claims			

D. ☐ Submission of Priority Documents

Priority document		Priority document	

E. ☒ Fees as specified on attached Fee Calculation sheet form PCT/RO/101 annex

IV. ☐ **A Request for Rectification under PCT 91** ☐ **A Petition** ☐ **A Sequence Listing Diskette**

V. ☐ **Other** (please specify):

The person signing this form is the:		
☐ Applicant	Robert Berliner	Typed name of signer
☒ Attorney/Agent (Reg. No.) 20,121		
☐ Common Representative	[signature]	Signature

PTO-1382 (Rev. 4-1995) (Modified)

U.S. Department of Commerce: Patent and Trademark Office

09/831843
PTO/PCT Rec'd 14 MAY 2001

TITLE: ENHANCED WAVEFORM INTERPOLATIVE CODER

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of Provisional Patent Application Nos. 60/110,522, filed December 1, 1998 and 60/110,641 filed December 1, 1998.

BACKGROUND OF THE INVENTION

Recently, there has been growing interest in developing toll-quality speech coders at rates of 4 kbps and below. The speech quality produced by waveform coders such as code-excited linear prediction (CELP) coders degrades rapidly at rates below 5 kbps [B. S. Atal, and M. R. Schroeder, "Stochastic Coding of Speech at Very Low Bit Rate", Proc. Int. Conf. Comm, Amsterdam, pp. 1610-1613, 1984]. On the other hand, parametric coders such as the waveform-interpolative (WI) coder, the sinusoidal-transform coder (STC), and the multiband-excitation (MBE) coder produce good quality at low rates, but they do not achieve toll quality [Y. Shoham, "High Quality Speech Coding at 2.4 to 4.0 kbps Based on Time Frequency-Interpolation", IEEE ICASSP'93, Vol. II, pp. 167-170, 1993; W. B. Kleijn, and J. Haagen, "Waveform Interpolation for Coding and Synthesis", in Speech Coding Synthesis by W. B. Kleijn and K. K. Paliwal, Elsevier Science B. V., Chapter 5, pp. 175-207, 1995; I. S. Burnett, and D. H. Pham, "Multi-Prototype Waveform Coding using Frame-by-Frame Analysis-by-Synthesis", IEEE ICASSP'97, pp. 1567-1570, 1997; R. J. McAulay, and T. F. Quatieri, "Sinusoidal Coding", in Speech Coding Synthesis by W. B. Kleijn and K. K. Paliwal, Elsevier Science B. V., Chapter 4, pp. 121-173, 1995; and D. Griffin, and J. S. Lim, "Multiband Excitation Vocoder", IEEE Trans. ASSP, Vol. 36, No. 8, pp. 1223-1235, August 1988]. This is mainly due to lack of robustness to parameter estimation, which is commonly done in open loop, and to inadequate modeling of non-stationary speech segments. Also, in parametric coders the phase information is commonly not transmitted, and this is for two reasons: first, the phase is of secondary perceptual significance; and second, no efficient phase quantization scheme is known. WI coders typically use a

fixed phase vector for the slowly evolving waveform [Shoham, *supra*; Kleijn et al, *supra*; and Burnett et al, *supra*]. For example, in Kleijn et al, a fixed male speaker extracted phase was used. On the other hand, waveform coders such as CELP, by directly quantizing the waveform, implicitly allocate an excessive number of bits to the phase information - more than is perceptually required.

SUMMARY OF THE INVENTION

The present invention overcomes the foregoing drawbacks by implementing a paradigm that incorporates analysis-by-synthesis (AbS) for parameter estimation, and a novel pitch search technique that is well suited for the non-stationary segments. In one embodiment, the invention provides a novel, efficient AbS vector quantization (VQ) encoding of the dispersion phase of the excitation signal to enhance the performance of the waveform interpolative (WI) coder at a very low bit-rate, which can be used for parametric coders as well as for waveform coders. The enhanced analysis-by-synthesis waveform interpolative (EWI) coder of this invention employs this scheme, which incorporates perceptual weighting and does not require any phase unwrapping.

The WI coders use non-ideal low-pass filters for downsampling and upsampling of the slowly evolving waveform (SEW). In another embodiment of the invention, A novel AbS SEW quantization scheme is provided, which takes the non-ideal filters into consideration. An improved match between reconstructed and original SEW is obtained, most notably in the transitions.

Pitch accuracy is crucial for high quality reproduced speech in WI coders. Still another embodiment of the invention provides a novel pitch search technique based on varying segment boundaries; it allows for locking onto the most probable pitch period during transitions or other segments with rapidly varying pitch.

Commonly in speech coding, the gain sequence is downsampled and interpolated. As a result it is often smeared during plosives and onsets. To alleviate this problem, a further embodiment of the invention provides a novel switched-predictive AbS gain VQ scheme based on temporal weighting.

More particularly, the invention provides a method for interpolative coding of input signals at low data rates in which there may be significant pitch transitivity,the signals having an evolving waveform, the method incorporating at least one, and preferably all, of the following steps:

(a) AbS VQ of the SEW whereby to reduce distortion in the signal by obtaining the accumulated weighted distortion between an original sequence of waveforms and a sequence of quantized and interpolated waveforms;

(b) AbS quantization of the dispersion phase;

(c) locking onto the most probable pitch period of the signal using both a spectral domain pitch search and a temporal domain pitch search;

(d) incorporating temporal weighting in the AbS VQ of the signal gain, whereby to emphasize local high energy events in the input signal;

(e) applying both high correlation and low correlation synthesis filters to a vector quantizer codebook in the AbS VQ of the signal gain whereby to add self correlation to the codebook vectors and maximize similarity between the signal waveform and a codebook waveform;

(f) using each value of gain in the AbS VQ of the signal gain to obtain a plurality of shapes, each composed of a predetermined number of values, and comparing said shapes to a vector quantized codebook of shapes, each having said predetermined number of values, e.g., in the range of 2 - 50, preferably 5 - 20; and

(g) using a coder in which a plurality of bits, e.g. 4 bits, are allocated to the SEW dispersion phase.

The method of the invention can be used in general with any waveform signal, and is particularly useful with speech signals. In the step of AbS VQ of the SEW, distortion is reduced in the signal by obtaining the accumulated weighted distortion between an original sequence of waveforms and a sequence of quantized and interpolated waveforms. In the step of AbS quantization of the dispersion phase, at least one codebook is provided that contains magnitude and phase information for predetermined waveforms. The linear phase of the input is crudely aligned, then iteratively shifted and compared to a plurality of waveforms reconstructed from the magnitude and

phase information contained in one or more codebooks. The reconstructed waveform that best matches one of the iteratively shifted inputs is selected.

In the step of locking onto the most probable pitch period of the signal, the invention includes searching the temporal domain pitch, defining a boundary for a segment of said temporal domain pitch, maximizing the length of the boundary by iteratively shrinking and expanding the segment, and maximizing the similarity by shifting the segment. The searches are preferably conducted respectively at 100 Hz and 500 Hz.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of the AbS SEW vector quantization;

Figure 2 shows amplitude-time plots illustrating the improved waveform matching obtained for a non-stationary speech segment by interpolating the optimized SEW;

Figure 3 is a block diagram of the AbS dispersion phase vector quantization;

Figure 4 is a plot of the segmentally weighted signal-to-noise ratio of the phase vector quantization versus the number of bits, for modified intermediate reference system (MIRS) and for non-MIRS (flat) speech;

Figure 5 shows the results of subjective A/B tests comparing a 4-bit phase vector quantization and a male extracted fixed phase;

Figure 6 is a block diagram of the pitch search of the EWI coder; and

Figure 7 is a block diagram of the switch-predictive AbS gain VQ using temporal weighting.

DETAILED DESCRIPTION OF THE INVENTION

The invention has a number of embodiments, some of which can be used independently of the others to enhance speech and other signal coding systems. The embodiments cooperate to produce a superior coding system, involving AbS SEW optimization, and novel dispersion phase quantizer, pitch search scheme, switched-predictive AbS gain VQ, and bit allocation.

AbS SEW Quantization

Commonly in WI coders the SEW is distorted by downsampling and upsampling with non-ideal low-pass filters. In order to reduce such distortion, an AbS SEW quantization scheme, illustrated in Figure 1, was used. Consider the accumulated weighted distortion, D_{wI}, between the input SEW vectors, $\mathbf{r}_m$, and the interpolated vectors, $\tilde{\mathbf{r}}_m$, given by:

$$D_{wI}(\hat{\mathbf{r}}_M, \{\mathbf{r}_m\}_{m=1}^{M+L-1}) = \begin{bmatrix} \sum_{m=1}^{M} [\mathbf{r}_m - \tilde{\mathbf{r}}_m]^H \mathbf{W}_m [\mathbf{r}_m - \tilde{\mathbf{r}}_m] \\ + \sum_{m=M+1}^{M+L-1} [1-\alpha(t_m)]^2 [\mathbf{r}_m - \tilde{\mathbf{r}}_M]^H \mathbf{W}_m [\mathbf{r}_m - \tilde{\mathbf{r}}_M] \end{bmatrix}$$

(1)

where the first sum is that of many current distortions and the second sum is that of lookahead distortions. H denotes Hermitian (transposed + complex conjugate), M is the number of waveforms per frame, L is the lookahead number of waveforms, $\alpha(t)$ is some increasing interpolation function in the range $0 \le \alpha(t) \le 1$, and $\mathbf{W}_m$ is a diagonal matrix whose elements, w_{kk}, are the combined spectral-weighting and synthesis of the k-th harmonic given by:

$$w_{kk} = \frac{1}{K} \left| \frac{gA(z/\gamma_1)}{\hat{A}(z)A(z/\gamma_2)} \right|^2_{z=e^{j(\frac{2\pi}{P})k}} \quad ; k = 1, \ldots, K \qquad (2)$$

where P is the pitch period, K is the number of harmonics, g is the gain, $A(z)$ and $\hat{A}(z)$ are the input and the quantized LPC polynomials respectively, and the spectral weighting parameters satisfy $0 \le \gamma_2 < \gamma_1 \le 1$. It is also possible to leave out the inverse of the number of harmonics, i.e., the 1/K parameter, the

gain, i.e. the g parameter, or another combination of input and quantized LPC polynomials, i.e. the $A(Z)$ and $\hat{A}(Z)$ parameters.

The interpolated SEW vectors are given by:

$$\tilde{\mathbf{r}}_m = [1-\alpha(t_m)]\hat{\mathbf{r}}_0 + \alpha(t_m)\hat{\mathbf{r}}_M \quad ; \quad m = 1, \ldots, M \tag{3}$$

where t is time, m is the number of waveforms in a frame, and $\hat{\mathbf{r}}_0$ and $\hat{\mathbf{r}}_M$ are the quantized SEW at the previous and at the current frame respectively. The parameter α is an increasing linear function from 0 to 1. It can be shown that the accumulated distortion in equation (1) is equal to the sum of modeling distortion and quantization distortion:

$$D_{wI}(\hat{\mathbf{r}}_M, \{\mathbf{r}_m\}_{m=1}^{M+L-1}) = D_{wI}(\mathbf{r}_{M,opt}, \{\mathbf{r}_m\}_{m=1}^{M+L-1}) + D_w(\hat{\mathbf{r}}_M, \mathbf{r}_{M,opt}) \tag{4}$$

where the quantization distortion is given by:

$$D_w(\hat{\mathbf{r}}_M, \mathbf{r}_{M,opt}) = (\hat{\mathbf{r}}_M - \mathbf{r}_{M,opt})^H \mathbf{W}_{M,opt}(\hat{\mathbf{r}}_M - \mathbf{r}_{M,opt}) \tag{5}$$

The optimal vector, $\mathbf{r}_{M,opt}$, which minimizes the modeling distortion, is given by:

$$\mathbf{r}_{M,opt} = \mathbf{W}_{M,opt}^{-1}\left[\begin{array}{l}\sum_{m=1}^{M}\alpha(t_m)\mathbf{W}_m\left[\mathbf{r}_m - [1-\alpha(t_m)]\hat{\mathbf{r}}_0\right] \\ + \sum_{m=M+1}^{M+L-1}[1-\alpha(t_m)]^2\mathbf{W}_m\mathbf{r}_m\end{array}\right] \tag{6}$$

where, $$\mathbf{W}_{M,opt} = \sum_{m=1}^{M}\alpha(t_m)^2\mathbf{W}_m + \sum_{m=M+1}^{M+L-1}[1-\alpha(t_m)]^2\mathbf{W}_m \tag{7}$$

Therefore, VQ with the accumulated distortion of equation (1) can be simplified by using the distortion of equation (5), and:

$$\hat{\mathbf{r}}_M = \underset{\mathbf{r}'_i}{\operatorname{argmin}}\left\{ (\mathbf{r}'_i - \mathbf{r}_{M,opt})^H \mathbf{W}_{M,opt}(\mathbf{r}'_i - \mathbf{r}_{M,opt}) \right\} \tag{6}$$

An improved match between reconstructed and original SEW is obtained, most notably in the transitions. Figure 2 illustrates the improved

waveform matching obtained for a non-stationary speech segment by interpolating the optimized SEW.

AbS Phase Quantization

The dispersion-phase vector quantization scheme is illustrated in Figure 3. Consider a pitch cycle which is extracted from the residual signal, and is cyclically shifted such that its pulse is located at position zero. Let its discrete Fourier transform (DFT) be denoted by r; the resulting DFT phase is the dispersion phase, φ , which determines, along with the magnitude $|\mathbf{r}|$, the waveform's pulse shape. The SEW waveform r is the vector of complex DFT coefficients. The complex number can represent magnitude and phase. After quantization, the components of the quantized magnitude vector, $|\hat{\mathbf{r}}|$, are multiplied by the exponential of the quantized phases, $\hat{\varphi}(k)$, to yield the quantized waveform DFT, $\hat{\mathbf{r}}$, which is subtracted from the input DFT to produce the error DFT. The error DFT is then transformed to the perceptual domain by weighting it by the combined synthesis and weighting filter W(z)/A(z). In a crude linear phase alignment, the encoder searches for the phase that minimizes the energy of the perceptual domain error, shifting the signal such that the peak is located at time zero. It then allows a refining cyclic shift of the input waveform during the search, incrementally increasing or decreasing the linear phase, to eliminate any residual phase shift between the input waveform and the quantized waveform. Although shown in Figure 3 as occurring immediately after the crude linear phase alignment, the refined linear phase alignment step can occur elsewhere in the cycle, e.g., between the X and + steps. Phase dispersion quantization aims to improve waveform matching. Efficient quantization can be obtained by using the perceptually weighted distortion:

$$D_w(\mathbf{r},\hat{\mathbf{r}}) = (\mathbf{r}-\hat{\mathbf{r}})^H \mathbf{W}(\mathbf{r}-\hat{\mathbf{r}}) \qquad (7)$$

The magnitude is perceptually more significant than the phase; and should therefore be quantized first. Furthermore, if the phase were quantized first, the very limited bit allocation available for the phase would lead to an excessively degraded spectral matching of the magnitude in favor of a

somewhat improved, but less important, matching of the waveform. For the above distortion, the quantized phase vector is given by:

$$\hat{\varphi} = \underset{\hat{\varphi}_i}{\operatorname{argmin}} \{ (\mathbf{r} - \mathbf{e}^{j\hat{\varphi}_i} |\hat{\mathbf{r}}|)^H \mathbf{W} (\mathbf{r} - \mathbf{e}^{j\hat{\varphi}_i} |\hat{\mathbf{r}}|) \} \tag{8}$$

where i is the running phase codebook index, and $\mathbf{e}^{j\hat{\varphi}_i}$ is the respective diagonal phase exponent matrix where i is the running phase codebook index, and the respective phase exponent matrix is given by

$$\mathbf{e}^{\hat{\varphi}j_i} = diagonal\left\{ e^{j\hat{\varphi}_i(k)} \right\}. \tag{9}$$

The AbS search for phase quantization is based on evaluating (8) for each candidate phase codevector. Since only trigonometric functions of the phase candidates are used, phase unwrapping is avoided. The EWI coder uses the optimized SEW, $\mathbf{r}_{M,opt}$, and the optimized weighting, $\mathbf{w}_{M,opt}$, for the AbS phase quantization.

$$\text{Equation (8)} = \underset{\hat{\varphi}_i}{\arg\max} \left\{ \int_0^{2\pi} r_W(\phi) \hat{r}_W(\hat{\varphi}_i, \phi) d\phi \right\}$$

Equivalently, the quantized phase vector can be simplified to:

$$\hat{\varphi} = \underset{\hat{\varphi}_i}{\arg\max} \left\{ \sum_{k=1}^{K} w_{kk} |r(k)| |\hat{r}(k)| \cos(\varphi(k) - \hat{\varphi}(k)_i) \right\} \tag{10}$$

where $\hat{\varphi}(k)$ is the phase of, r(k), the k-th input DFT coefficient. The average global distortion measure for M vector set is:

$$D_{w,Global} = \frac{1}{M} \sum_{m=\{Data\ Vectors\}} D_W(r_m, \mathbf{e}^{j\hat{\varphi}_m} |\hat{r}|_m)$$

$$= \frac{1}{M} \sum_{m=\{Data\ Vectors\}} \frac{1}{K_m} \sum_{k=1}^{K_m} w_{kk,m} \left| r(k)_m - e^{j\hat{\varphi}(k)_m} |\hat{r}(k)|_m \right|^2 \tag{11}$$

The centroid equation [A. Gersho et al, "Vector Quantization and Signal Compression", Kluwer Academic Publishers, 1992] of the k-th harmonic's phase for the j-th cluster, which minimizes the global distortion in equation (11), is given by:

$$\hat{\varphi}(k)_{j^{th}-cluster} = \operatorname{atan}\left[\frac{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|\hat{r}(k)_m\right| \left|r(k)_m\right| \sin(\varphi(k)_m)}{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|\hat{r}(k)_m\right| \left|r(k)_m\right| \cos(\varphi(k)_m)}\right]$$

These centroid equations use trigonometric functions of the phase, and therefore do not require any phase unwarping. It is possible to use $\left|r(k)_m\right|^2$ instead of $\left|\hat{r}(k)_m\right|\left|r(k)_m\right|$.

The phase vector's dimension depends on the pitch period and, therefore, a variable dimension VQ has been implemented. In the WI system the possible pitch period value was divided into eight ranges, and for each range of pitch period an optimal codebook was designed such that vectors of dimension smaller than the largest pitch period in each range are zero padded.

Pitch changes over time cause the quantizer to switch among the pitch-range codebooks. In order to achieve smooth phase variations whenever such switch occurs, overlapped training clusters were used.

The phase-quantization scheme has been implemented as a part of WI coder, and used to quantize the SEW phase. The objective performance of the suggested phase VQ has been tested under the following conditions:

- Phase Bits: 0-6 every 20ms, a bitrate of 0-300 bit/second.
- 8 pitch ranges were selected, and training has been performed for each range.
- Modified IRS (MIRS) filtered speech (Female+Male)
    - Training Set: 99,323 vectors.
    - Test Set: 83,099 vectors.
- Non-MIRS filtered speech (Female+Male)

    - Training Set: 101,359 vectors.
    - Test Set: 95,446 vectors.
- The magnitude was not quantized.

The segmental weighted signal-to-noise ratio (SNR) of the quantizer is illustrated in Figure 4. The proposed system achieves approximately 14dB SNR for as low as 6 bits for non-MIRS filtered speech, and nearly 10dB for MIRS filtered speech.

Recent WI coders have used a male speaker extracted dispersion phase [Kleijn et al, *supra*; Y. Shoham, "Very Low Complexity Interpolative Speech Coding at 1.2 to 2.4 KBPS", IEEE ICASSP '97, pp. 1599-1602, 1997].A subjective A/B test was conducted to compare the dispersion phase of this invention, using only 4 bits, to a male extracted dispersion phase. The test data included 16 MIRS speech sentences, 8 of which are of female speakers, and 8 of male speakers. During the test, all pairs of file were played twice in alternating order, and the listeners could vote for either of the systems, or for no preference. The speech material was synthesized using WI system in which only the dispersion phase was quantized every 20ms. Twenty one listeners participated in the test. The test results, illustrated in Figure 5, show improvement in speech quality by using the 4-bit phase VQ. The improvement is larger for female speakers than for male. This may be explained by a higher number of bits per vector sample for female, by less spectral masking for female's speech, and by a larger amount of phase-dispersion variation for female. The codebook design for the dispersion-phase quantization involves a tradeoff between robustness in terms of smooth phase variations and waveform matching. Locally optimized codebook for each pitch value may improve the waveform matching on the average, but may occasionally yield abrupt and excessive changes which may cause temporal artifacts.

Pitch Search

The pitch search of the EWI coder consists of a spectral domain search employed at 100 Hz and a temporal domain search employed at 500

Hz, as illustrated in Figure 6. The spectral domain pitch search is based on harmonic matching [McAuley et al, *supra*; Griffin et al, *supra*; and E. Shlomot, V. Cuperman, and A. Gersho, "Hybrid Coding of Speech at 4 kbps", IEEE Speech Coding Workshop, pp. 37-38, 1997]. The temporal domain pitch search is based on varying segment boundaries. It allows for locking onto the most probable pitch period even during transitions or other segments with rapidly varying pitch (e.g., speech onset or offset or fast changing periodicity). Initially, pitch periods, $P(n_i)$, are searched every 2 ms at instances n_i by maximizing the normalized correlation of the weighted speech $s_w(n)$, that is:

$$P(n_i) = \underset{\tau, N_1, N_2}{\arg\max} \left\{ \rho(n_i, \tau, N_1, N_2) \right\} =$$

$$\underset{\tau, N_1, N_2}{\arg\max} \left\{ \frac{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n) s_w(n-\tau)}{\sqrt{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n) s_w(n)} \sqrt{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n-\tau) s_w(n-\tau)}} \right\} \quad (12)$$

where τ is the shift in the segment, Δ is some incremental segment used in the summations for computational simplicity, and $0 \le N_j \le \lfloor 160/\Delta \rfloor$. Then, every 10 ms a weighted-mean pitch value is calculated by:

$$P_{mean} = \sum_{i=1}^{5} \rho(n_i) P(n_i) / \sum_{i=1}^{5} \rho(n_i) \quad (13)$$

where $\rho(n_i)$ is the normalized correlation for $P(n_i)$. The above values (160, 10, 5) are for the particular coder and is used for illustration.

Equation (12) describes the temporal domain pitch search and the temporal domain pitch refinement blocks of Figure 6. Equation (13) describes the weighted average pitch block of Figure 6.

Gain Quantization

The gain trajectory is commonly smeared during plosives and onsets by downsampling and interpolation. This problem is addressed and speech crispness is improved in accordance with an embodiment of the invention

that provides a novel switched-predictive AbS gain VQ technique, illustrated in Figure 7. Switched-prediction is introduced to allow for different levels of gain correlation, and to reduce the occurrence of gain outliers. In order to improve speech crispness, especially for plosives and onsets, temporal weighting is incorporated in the AbS gain VQ. The weighting is a monotonic function of the temporal gain. Two codebooks of 32 vectors each are used. Each codebook has an associated predictor coefficient, P_i, and a DC offset D_i. The quantization target vector is the DC removed log-gain vector denoted by $t(m)$. The search for the minimal weighted mean squared error (WMSE) is performed over all the vectors, $c_{ij}(m)$, of the codebooks. The quantized target, $\hat{t}(m)$, is obtained by passing the quantized vector, $c_{ij}(m)$, through the synthesis filter. Since each quantized target vector may have a different value of the removed DC, the quantized DC is added temporarily to the filter memory after the state update, and the next quantized vector's DC is subtracted from it before filtering is performed. Since the predictor coefficients are known, direct VQ can be used to simplify the computations. The synthesis filter adds self correlation to the codebook vector. All combinations are tried and whether high or low self correlation is used depends on which yields the best results.

Bit Allocation

The bit allocation of the coder is given in Table 1. The frame length is 20 ms, and ten waveforms are extracted per frame. The pitch and the gain are coded twice per frame.

Table 1- Bit allocation for EWI coder

Parameter	Bits / Frame	Bits / second
LPC	18	900
Pitch	2x6=12	600
Gain	2x6=12	600
REW	20	1000
SEW magn.	14	700
SEW phase	4	200

Total	80	4000

Subjective Results

A subjective A/B test was conducted to compare the 4 kbps EWI coder of this invention to MPEG-4 at 4 kbps, and to G.723.1. The test data included 24 MIRS speech sentences, 12 of which are of female speakers, and 12 of male speakers. Fourteen listeners participated in the test. The test results, listed in Tables 2 to 4, indicate that the subjective quality of EWI exceeds that of MPEG-4 at 4 kbps and of G.723.1 at 5.3 kbps, and it is slightly better than that of G.723.1 at 6.3 kbps.

Table 2

Test	4 kbps WI	4 kbps MPEG-4
Female	65.48%	34.52%
Male	61.90%	38.10%
Total	63.69%	36.31%

Table 2 shows the results of subjective A/B tests for comparison between the 4 kbps WI coder and th 4 kbps MPEG-4. With 95% certainty the WI preference lies in [58.63%, 68.75%].

Table 3

Test	4 kbps WI	5.3 kbps G.723.1
Female	57.74%	42.26%
Male	61.31%	38.69%
Total	59.52%	40.48%

Table 3 shows the results of subjective A/B tests for comparison between the 4 kbps WI coder to 5.3 kbps G.723.1. With 95% certainty the WI preference lies in [54.17%, 64.88%]

Table 4

Test	4 kbps WI	6.3 kbps G.723.1
Female	54.76%	45.24%
Male	52.98%	47.02%
Total	53.87%	46.13%

Table 4. Results of subjective A/B test for comparison between the 4 kbps WI coder to 6.3 kbps G.723.1. With 95% certainty the WI preference lies in [48.51%, 59.23%].

The present invention incorporates several new techniques that enhance the performance of the WI coder, analysis-by-synthesis vector-quantization of the dispersion-phase, AbS optimization of the SEW, a special pitch search for transitions, and switched-predictive analysis-by-synthesis gain VQ. These features improve the algorithm and its robustness. The test results indicate that the performance of the EWI coder slightly exceeds that of G.723.1 at 6.3 kbps and therefore EWI achieves very close to toll quality, at least under clean speech conditions.

THE CLAIMS

1. A method for interpolative coding input signals at low data rates in which there is significant pitch transitivity, and wherein said signals said signals may have a slowly evolving waveform, the method incorporating at least one of the following steps:

(a) analysis-by-synthesis vector-quantization of the slowly evolving waveform;

(b) analysis-by-synthesis quantization of the dispersion phase;

(c) locking onto the most probable pitch period of the signal using both a spectral domain pitch search and a temporal domain pitch search;

(d) incorporating temporal weighting in the analysis-by-synthesis vector-quantization of the signal gain;

(e) applying both high correlation and low correlation synthesis filters to a vector quantizer codebook in the analysis-by-synthesis vector-quantization of the signal gain whereby to add self correlation to the codebook vectors;

(f) using each value of gain in the analysis-by-synthesis vector-quantization of the signal gain; and

(g) using a coder in which a plurality of bits therein are allocated to the slowly evolving waveform phase.

2. The method of claim 1 in which said signal is speech.

3. The method of claim 1 in which said method incorporates each of steps (a) through (g).

4. The method of claim 1 in which in the step of analysis-by-synthesis vector-quantization of the slowly evolving waveform, distortion is reduced in the signal by obtaining the accumulated weighted distortion between an original sequence of waveforms and a sequence of quantized and interpolated waveforms.

5. The method of claim 1 including providing at least one codebook containing magnitude and phase information for predetermined

waveforms, and in which the step of analysis-by-synthesis quantization of the dispersion phase is conducted by crudely aligning the linear phase of the input, then iteratively shifting said crudely aligned linear phase input, comparing the shifted input to a plurality of waveforms reconstructed from the magnitude and phase information contained in said at least one codebook, and selecting the reconstructed waveform that best matches one of the iteratively shifted inputs.

6. The method of claim 1 in which in the method of searching the temporal domain pitch in said step of locking onto the most probable pitch period of the signal, comprises defining a boundary for a segment of said temporal domain pitch, selecting the best boundary and maximizing the similarity by iteratively shifting the segment, and by shrinking and expanding the segment,

7. The method of claim 1 in which the spectral domain pitch and temporal domain pitch searches, in said step of locking onto the most probable pitch period of the signals, are conducted respectively at 100 Hz and 500 Hz.

8. The method of claim 1 in which the step of the temporal weighting in the analysis-by-synthesis vector-quantization of the signal gain is changed as a function of time whereby to emphasize local high energy events in the input signal.

9. The method of claim 1 in which selection between the high and low correlation synthesis filters in the analysis-by-synthesis vector-quantization of the signal gain is made to maximize similarity between the gain waveform and a codebook waveform.

10. The method of claim 1 wherein each value of gain in the analysis-by-synthesis vector-quantization of the signal gain is used to obtain a plurality of shapes, each composed of a predetermined number of values, and

comparing said shapes to a vector quantized codebook of shapes, each having said predetermined number of values.

11. A method for interpolative coding input signals at low data rates in which said signals have a slowly evolving waveform, the method incorporating analysis-by-synthesis vector-quantization of the slowly evolving waveform.

12. The method of claim 11 in which distortion is reduced in the signal by obtaining the accumulated weighted distortion between an original sequence of waveforms and a sequence of quantized and interpolated waveforms.

13. A method for interpolative coding input signals at low data speeds in which the signal has a slowly evolving waveform having a dispersion phase, the method incorporating analysis-by-synthesis quantization of the dispersion phase.

14. The method of claim 13 including providing at least one codebook containing magnitude and phase information for predetermined waveforms, crudely aligning the linear phase of the input, then iteratively shifting said crudely aligned linear phase input, comparing the shifted input to a plurality of waveforms reconstructed from the magnitude and phase information contained in said at least one codebook, and selecting the reconstructed waveform that best matches one of the iteratively shifted inputs.

15. The method of claim 14 in which the average global distortion measure for a particular vector set M is:

$$\frac{1}{M} \sum_{\substack{m=\{Data \\ Vectors\}}} \frac{1}{K_m} \sum_{k=1}^{K_m} w_{kk,m} \left| r(k)_m - e^{j\hat{\varphi}(k)_m} \left|\hat{r}(k)\right|_m \right|^2$$

and including the step of minimizing the global distortion thereof by using the following formula for the k-th harmonic's phase for the j-th cluster:

$$\hat{\varphi}(k)_{j^{th}-cluster} = \operatorname{atan}\left[\frac{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|r(k)_m\right|^2 \sin(\varphi(k)_m)}{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|r(k)_m\right|^2 \cos(\varphi(k)_m)}\right]$$

16. The method of claim 14 in which the average global distortion measure for a particular vector set M is:

$$\frac{1}{M} \sum_{m=\{Data\ Vectors\}} \frac{1}{K_m} \sum_{k=1}^{K_m} w_{kk,m} \left| r(k)_m - e^{j\hat{\varphi}(k)_m} \left|\hat{r}(k)\right|_m \right|^2$$

and including the step of minimizing the global distortion thereof by using the following formula for the k-th harmonic's phase for the j-th cluster:

$$\hat{\varphi}(k)_{j^{th}-cluster} = \operatorname{atan}\left[\frac{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|\hat{r}(k)_m\right| \left|r(k)_m\right| \sin(\varphi(k)_m)}{\sum\limits_{m=\{j^{th}-cluster\}} \frac{1}{K_m} w_{kk,m} \left|\hat{r}(k)_m\right| \left|r(k)_m\right| \cos(\varphi(k)_m)}\right]$$

17. A method for interpolative coding input signals at low data rates, comprising locking onto the most probable pitch period of the signal using both a spectral domain pitch search and a temporal domain pitch search.

18. The method of claim 17 in which in the method of searching the temporal domain pitch comprises defining a boundary for a segment of said temporal domain pitch, selecting the location of the boundaries that

maximize the similarity by iteratively shrinking and expanding the segment and by shifting the segment.

19. The method of claim 18 in which the method of searching the temporal domain pitch is in accordance with the formula:

$$P(n_i) = \underset{\tau, N_1, N_2}{\arg\max} \left\{ \rho(n_i, \tau, N_1, N_2) \right\} =$$

$$\underset{\tau, N_1, N_2}{\arg\max} \left\{ \frac{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n) s_w(n-\tau)}{\sqrt{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n) s_w(n)} \sqrt{\sum_{n=n_i-N_1\Delta}^{n_i+\tau+N_2\Delta} s_w(n-\tau) s_w(n-\tau)}} \right\}$$

where τ is the shift in the segment, Δ is some incremental segment used in the summations for computational simplicity, and N_j is a number calculated for the coder.

20. The method of claim 19 including the step of obtaining the weighted average pitch in accordance with the formula:

$$P_{mean} = \sum_{i=1}^{5} \rho(n_i) P(n_i) / \sum_{i=1}^{5} \rho(n_i)$$

where $\rho(n_i)$ is the normalized correlation for $P(n_i)$.

21. The method of claim 19 in which the spectral domain pitch and temporal domain pitch searches in said step of locking onto the most probable pitch period of the signals are conducted respectively at 100 Hz and 500 Hz.

22. A method for interpolative coding input signals at low data speeds, comprising incorporating temporal weighting in the analysis-by-synthesis vector-quantization of the signal gain.

23. The method of claim 22 in which the temporal weighting is changed as a function of time whereby to emphasize local high energy events in the input signal.

24. A method for interpolative coding input signals at low data speeds, comprising applying both high correlation and low correlation synthesis filters to a vector quantizer codebook in the analysis-by-synthesis vector-quantization of the signal gain whereby to add self correlation to the codebook vectors.

25. The method of claim 24 in which selection between the high and low correlation synthesis filters is made to maximize similarity between the signal waveform and a codebook waveform.

26. A method for interpolative coding input signals at low data speeds, comprising using each value of gain in the analysis-by-synthesis vector-quantization of the signal gain.

27. The method of claim 26 wherein each value of gain is used to obtain a plurality of shapes, each composed of a predetermined number of values, and comparing said shapes to a vector quantized codebook of shapes, each having said predetermined number of values.

28. The method of claim 27 in which said predetermined number of values is in the range of 2 to 50.

29. The method of claim 28 in which said predetermined number of values is in the range of 5 to 20.

30. A method for interpolative coding input signals at low data speeds in which said signals have a slowly evolving waveform, comprising using a coder in which a plurality of bits therein are allocated to the slowly evolving waveform phase.

31. The method of claim 30 in which 4 bits are allocated to the slowly evolving waveform phase in the coder.

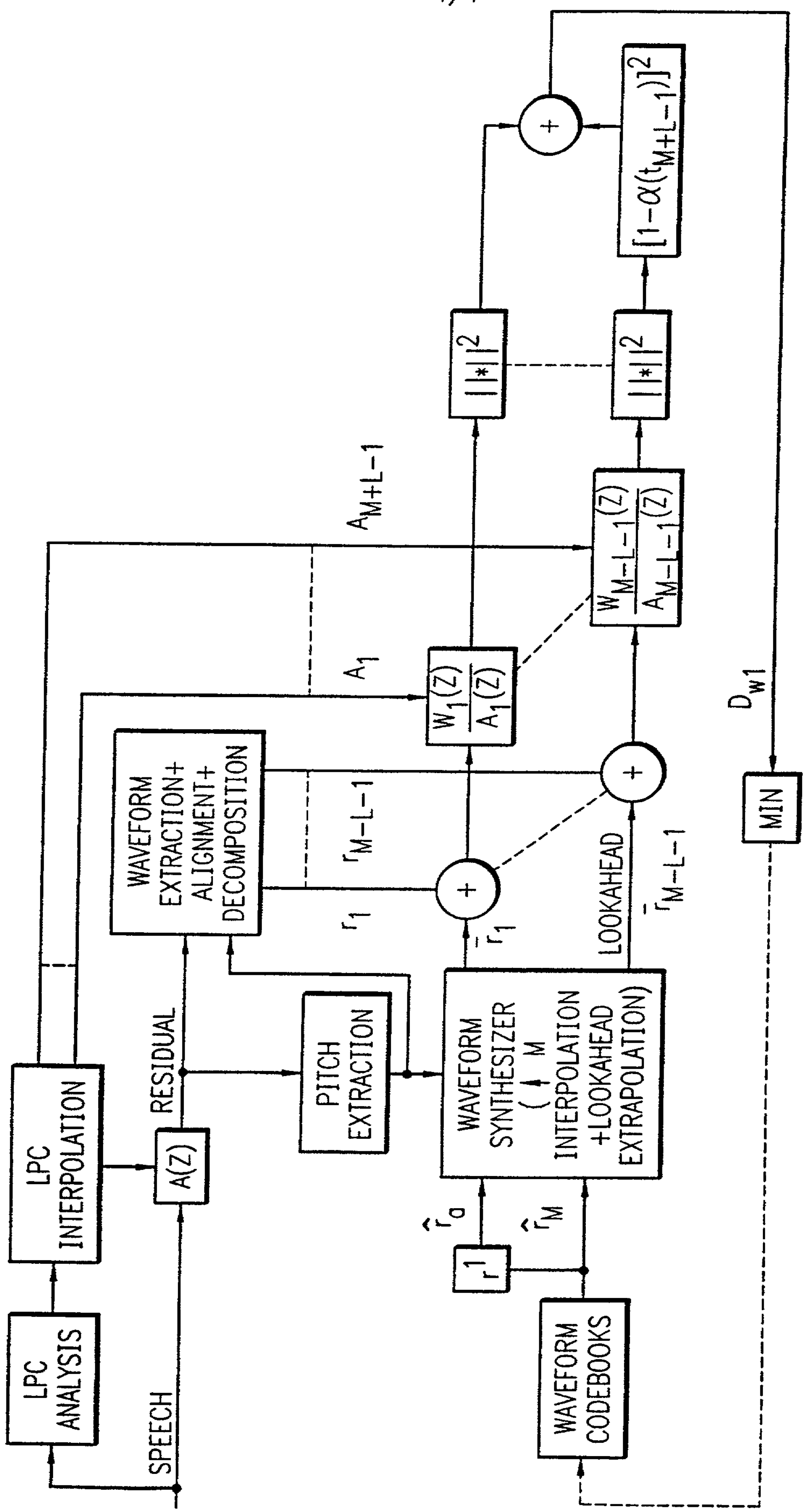

LPC ANALYSIS
LPC INTERPOLATION
SPEECH
A(Z)
RESIDUAL
PITCH EXTRACTION
WAVEFORM EXTRACTION+ ALIGNMENT+ DECOMPOSITION
WAVEFORM SYNTHESIZER (1 → M INTERPOLATION +LOOKAHEAD EXTRAPOLATION)
WAVEFORM CODEBOOKS
z^{-1}
$\hat{r}_a$
$\hat{r}_M$
r_1
$\bar{r}_1$
r_{M-L-1}
$\bar{r}_{M-L-1}$
LOOKAHEAD
A_1
A_{M+L-1}
$\frac{W_1(Z)}{A_1(Z)}$
$\frac{W_{M-L-1}(Z)}{A_{M-L-1}(Z)}$
$||*||^2$
$||*||^2$
$[1-\alpha(t_{M+L-1})]^2$
D_{w1}
MIN

FIG. 1

2/4

FIG. 2

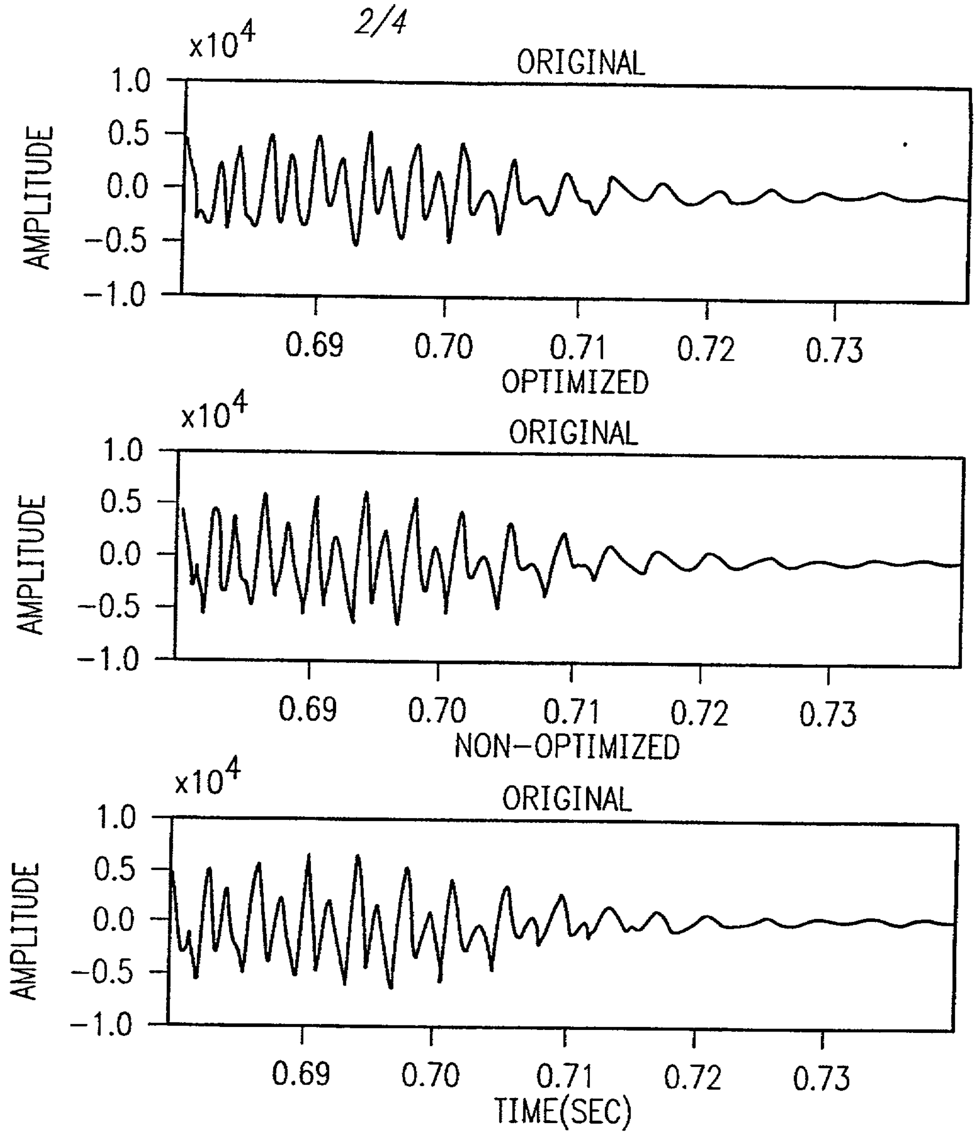

x10^4
ORIGINAL
1.0
0.5
0.0
−0.5
−1.0
AMPLITUDE
0.69
0.70
0.71
0.72
0.73
OPTIMIZED
x10^4
ORIGINAL
1.0
0.5
0.0
−0.5
−1.0
AMPLITUDE
0.69
0.70
0.71
0.72
0.73
NON−OPTIMIZED
x10^4
ORIGINAL
1.0
0.5
0.0
−0.5
−1.0
AMPLITUDE
0.69
0.70
0.71
0.72
0.73
TIME(SEC)

FIG. 3

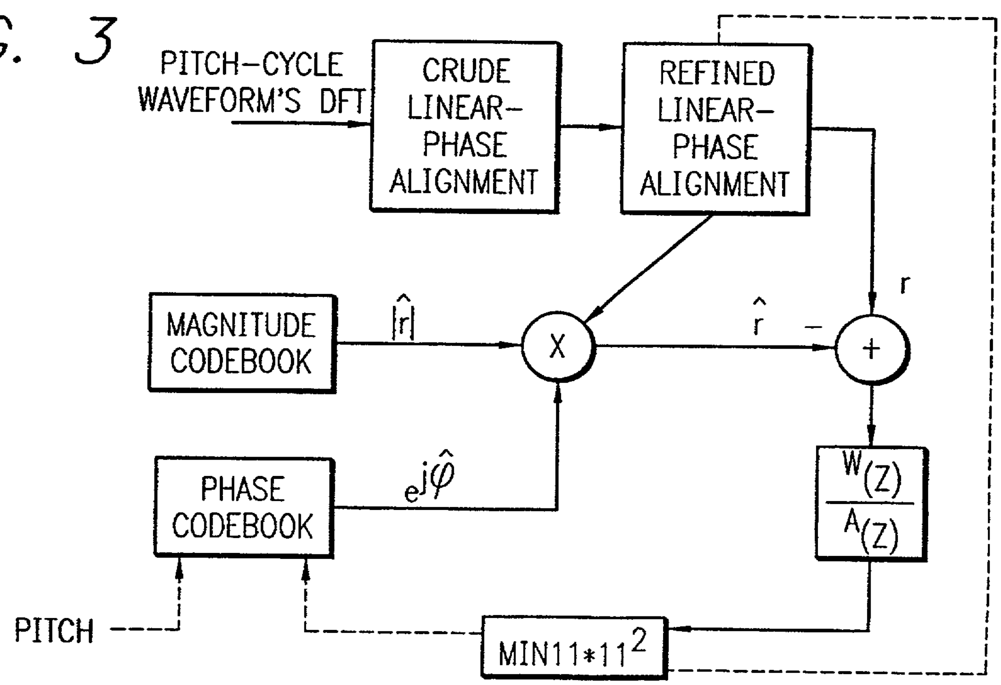

PITCH−CYCLE WAVEFORM'S DFT
CRUDE LINEAR−PHASE ALIGNMENT
REFINED LINEAR−PHASE ALIGNMENT
MAGNITUDE CODEBOOK
|r̂|
X
r̂
−
+
r
PHASE CODEBOOK
$e^{j\hat{\varphi}}$
W(Z)/A(Z)
PITCH
MIN11*11^2

FIG. 4

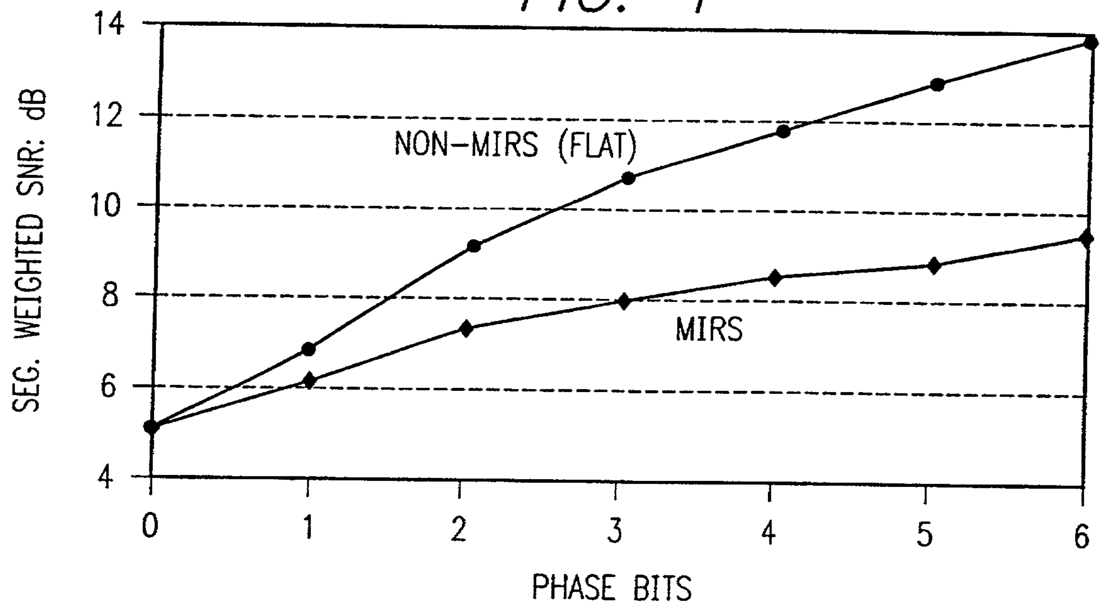
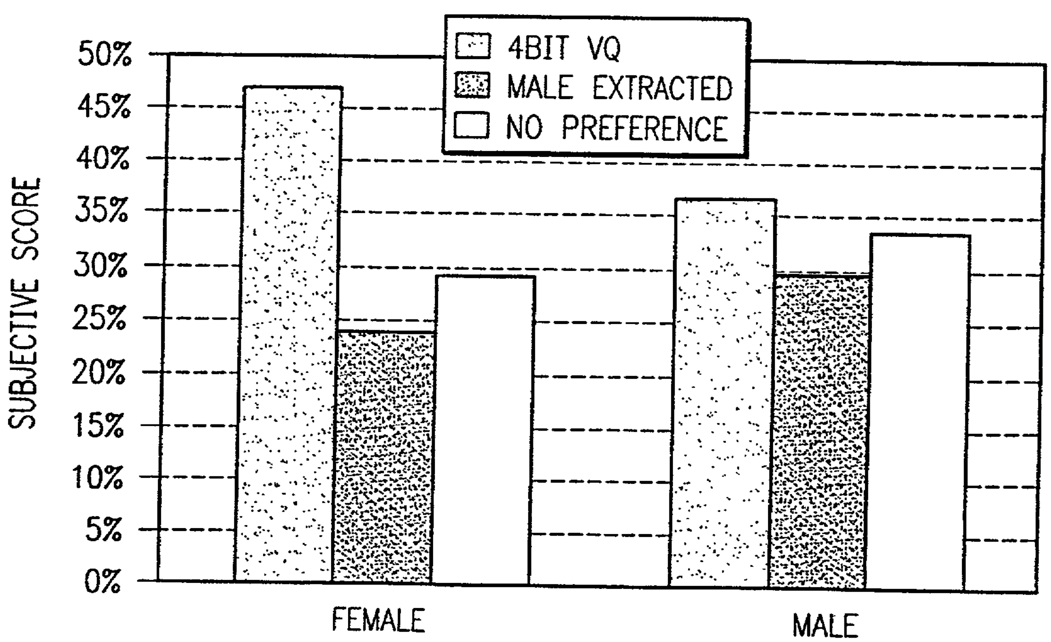

14
12
10
8
6
4
SEG. WEIGHTED SNR: dB
NON-MIRS (FLAT)
MIRS
0
1
2
3
4
5
6
PHASE BITS

FIG. 5


4BIT VQ
MALE EXTRACTED
NO PREFERENCE
50%
45%
40%
35%
30%
25%
20%
15%
10%
5%
0%
SUBJECTIVE SCORE
FEMALE
MALE

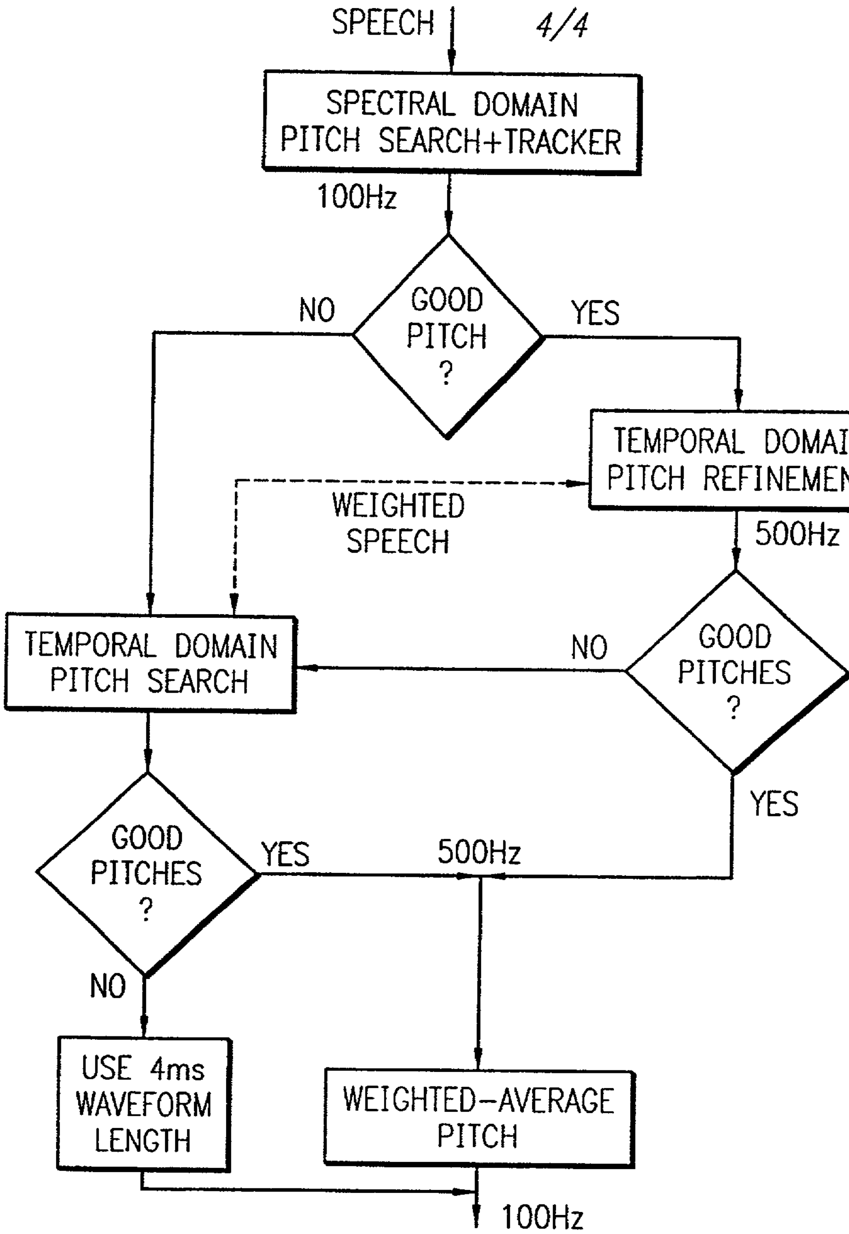

SPEECH
SPECTRAL DOMAIN PITCH SEARCH+TRACKER
100Hz
GOOD PITCH ?
NO
YES
TEMPORAL DOMAIN PITCH REFINEMENT
WEIGHTED SPEECH
500Hz
TEMPORAL DOMAIN PITCH SEARCH
GOOD PITCHES ?
NO
YES
GOOD PITCHES ?
YES
500Hz
NO
USE 4ms WAVEFORM LENGTH
WEIGHTED-AVERAGE PITCH
100Hz

FIG. 6

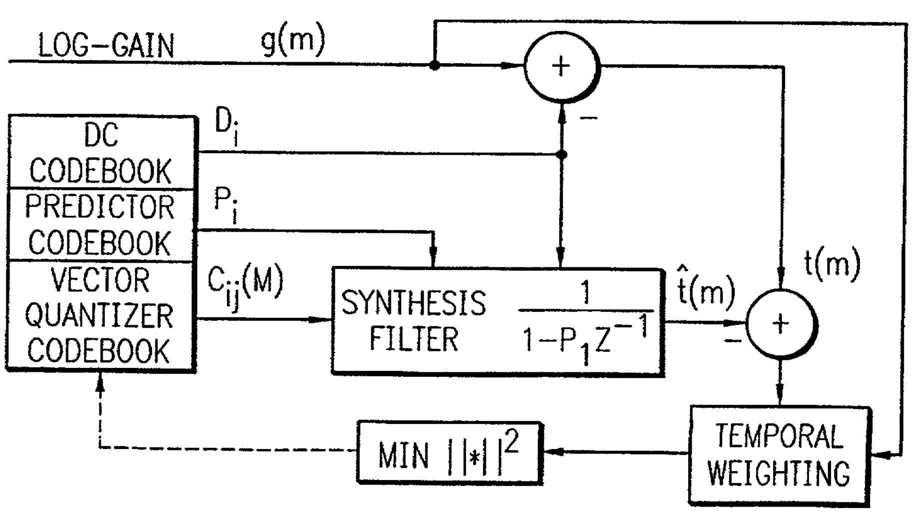

LOG-GAIN
g(m)
DC CODEBOOK
PREDICTOR CODEBOOK
VECTOR QUANTIZER CODEBOOK
D_i
P_i
$C_{ij}(M)$
SYNTHESIS FILTER
$\frac{1}{1-P_1Z^{-1}}$
$\hat{t}(m)$
t(m)
+
−
TEMPORAL WEIGHTING
MIN $||*||^2$

FIG. 7

DECLARATION AND PETITION

As the below named invêntors, I hereby declare that:

My residences post office address and citizenship are as stated below next to my name.

I believe that I am the original, first inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled ENHANCED WAVEFORM INTERPOLATIVE CODER, the specification of which was filed on December 10, 1999 and was assigned International Application No. PCT/US99/28449 and filed under 35 U.S.C. 371 on **May 14, 2001,** and assigned Serial No. **09/831,843.**

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, § 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, § 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed: **NONE**

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, § 112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, § 1.56(a) which occurred

between the filing date of the prior application and the national or PCT international filing date of this application: International Application No. **PCT/US99/28449 filed December 1, 1999.**

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Wherefore I pray that Letters Patent be granted to me for the invention or discovery described and claimed in the foregoing specification and claims, and I hereby subscribe my name to the foregoing specification and claims, declaration and petition.

Full name of sole or first inventor: ODED GOTTESMAN

Inventor's signature: Oded Gottesman

Date: July 10, 2001

Residence: 791-D LAUREL WALK
Goleta CA 93117

Citizenship: (~~United States of America~~) ISRAEL, Permanent Resident of USA
Post Office Address: same

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